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PATENTS ACT 1977

PATENTS FORM NO. 1/77 (R[illegible]d 1982)

(Rules 16, 19)

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Form 1/77 P1614

THE PATENT OFFICE
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19 MAY 2000
NEWPORT

REQUEST FOR GRANT OF A PATENT **0011973.5** 19 MAY 2000

THE GRANT OF A PATENT IS REQUESTED BY THE UNDERSIGNED ON THE BASIS OF THE PRESENT APPLICATION

I Applicant's or Agent's reference *(Please insert if available)* P1614

II Title of invention Steerable antennae

III Applicant or Applicants *(See note 2)*

Name (First or only applicant) 1... IPR Limited

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(a) The applicant(s) is/are the sole/joint inventor(s)
or
(b) A statement on Patents Form No 7/77 is/will be furnished

V Name of Agent (if any) *(See note 4)* J P L Hooper ADP CODE NO ~~06273020001~~

VI Address for Service *(See note 5)*
5 Haslingfield Road, Harlton, Cambridge CB3 7ER 402302 4001

VII Declaration of Priority *(See note 6)*

Country	Filing date	File number

VIII The Application claims an earlier date under Section 8(3), 12(6), 15(4), or 37(4) *(See note 7)*

Earlier application or patent number .... and filing date ....

IX Check List *(To be filled in by applicant or agent)*

A The application contains the following number of sheet(s)

1 Request ...... 1 .... Sheet(s)

2 Description .... 34 .... Sheet(s)

3 Claim(s) ...... No .... Sheet(s)

4 Drawing(s) .... 11 + 11 .... Sheet(s)

5 Abstract ...... No .... Sheet(s)

B The application as filed is accompanied by:-

1 Priority document . N/A

Translation of priority document N/A

3 Request for Search ...... No . .

4 Statement of Inventorship and Right to Grant .... No . .

X It is suggested that Figure No....... 1 .... of the drawings (if any) should accompany the abstract when published.

XI Signature *(See note 8)*

(J P L Hooper) Agent for the Applicants

NOTES:

1. This form, when completed, should be brought or sent to the Patent Office together with the prescribed fee and two copies of the description of the invention, and of any drawings.

2. Enter the name and address of each applicant. Names of individuals should be indicated in full and the surname or family name should be underlined. The names of all partners in a firm must be given in full. Bodies corporate should be designated by their corporate name and the country of incorporation and, where appropriate, the state of incorporation within that country should be entered where provided. Full corporate details, eg a "corporation organised and existing under the laws of the State of Delaware, United States of America", trading styles, eg "trading as xyz company", nationality, and former names, eg "formerly (known as) ABC Ltd" are *not* required and should *not* be given. Also enter applicant(s) ADP Code No.(if known).

3. Where the applicant or applicants is/are the sole inventor or the joint inventors, the declaration (a) to that effect at IV should be completed, and the alternative statement (b) deleted. If, however, this is not the case the declaration (a) should be struck out and a statement will then be required to be filed upon Patent Form No 7/77.

4. If the applicant has appointed an agent to act on his behalf, the agent's name and the address of his place of business should be indicated in the spaces available at V and VI. Also insert agent's ADP Code No. (if known) in the box provided.

5. An address for service in the United Kingdom to which all documents may be sent must be stated at VI. It is recommended that a telephone number be provided if an agent is not appointed.

6. The declaration of priority at VII should state the date of the previous filing and the country in which it was made and indicate the file number, if available.

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Steerable antennae

This invention relates to steerable antennae, and concerns in particular digital electronically-steerable acoustic antennae.

Phased array antennae are well known in the art in both the electromagnetic and the ultrasonic acoustic fields. They are less well known, but exist in simple forms, in the sonic (audible) acoustic area. These latter are relatively crude, and the invention seeks to provide a much superior audible acoustic array capable of being steered so as to direct its output more or less at will.

More specifically, the invention proposes in one aspect a fully digital steerable sonic/audible acoustic phased array antenna (a Digital Phased-Array Antennae, or DPAA) system comprising a plurality of spatially-distributed sonic electroacoustic transducers (SETs) arranged in a two-dimensional array and each connected to a single digital signal input *via* an input signal Distributor which modifies the input signal prior to feeding it to each SET in order to achieve the desired directional effect.

The description and Figures provided hereinafter necessarily describe the invention using block diagrams, with each block representing a hardware component or a signal processing step. The invention could, in principle, be realised by building separate physical components to perform each step, and interconnecting

them as shown. Several of the steps could be implemented using dedicated or programmable integrated circuits, possibly combining several steps in one circuit. It will be understood that in practice it is likely to be most convenient to perform several of the signal processing steps in software, using Digital Signal Processors (DSPs) or general purpose microprocessors. Sequences of steps could then be performed by separate processors or by separate software routines sharing a microprocessor, or be combined into a single routine to improve efficiency.

The various possibilities inherent in this, and the versions that are actually preferred, will be seen from the following:-

The SETs are preferably arranged in a plane or curved surface (a Surface), rather than randomly in space. They may also, however, be in the form of a 3-dimensional stack of two or more adjacent sub-arrays - two or more closely-spaced parallel plane or curved surfaces located one behind the next.

Within a Surface the SETs making up the array are preferably closely spaced, and ideally completely fill the overall antenna aperture. This is impractical with real circular-section SETs but may be achieved with triangular, square or hexagonal section SETs, or in general with any section which tiles the plane. Where the SET sections do not tile the plane, a close approximation to a filled aperture may be achieved by making the array in the form of a stack of arrays - ie, three-dimensional - where at least one additional Surface of SETs is mounted behind at least one other

such Surface, and the SETs in the or each rearward array radiate between the gaps in the frontward array(s).

The SETs are preferably similar, and ideally they are identical. They are, of course, sonic - that is, audio - devices, and most preferably they are able uniformly to cover the entire audio band from perhaps as low as (or lower than) 20Hz, to as much as 20KHz or more (the Audio Band). Alternatively, there can be used SETs of different sonic capabilities but together covering the entire range desired. Thus, multiple different SETs may be physically grouped together to form a composite SET (CSET) wherein the groups of different SETs together can cover the Audio Band even though the individual SETs cannot. As a further variant, SETs each capable of only partial Audio Band coverage can be not grouped but instead scattered throughout the array with enough variation amongst the SETs that the array as a whole has complete or more nearly complete coverage of the Audio Band.

An alternative form of CSET contains several (typically two) identical transducers, each driven by the same signal. This reduces the complexity of the required signal processing and drive electronics while retaining many of the advantages of a large DPAA.

Within a Surface the spacing of the SETs or CSET (hereinafter the two are denoted just by SETs) - that is, the general layout and structure of the array and the way the individual transducers are disposed therein - is preferably regular, and their distribution about the Surface is desirably symmetrical. Thus, the SETs are most preferably spaced in a triangular, square or hexagonal lattice. The type and orientation of the lattice can be chosen to control the spacing and direction of side-lobes.

Though not essential, each SET is very preferably omnidirectional in at least a hemisphere at all sound wavelengths which it is capable of effectively radiating (or receiving).

Each SET may take any convenient or desired form of sound radiating device (loudspeaker), and though they are all preferably the same they could be different. Most preferably they are loudspeakers of the type known as *pistonic* acoustic radiators (wherein the transducer diaphragm bodily moves a piston). And in such a case the maximum radiating extent of the piston-radiators (e.g., the effective piston diameter for circular SETs) of the individual SETs is preferably as small as possible, and ideally is as small as or smaller than the acoustic wavelength of the highest frequency in the Audio Band (e.g. in air, 20KHz sound waves have a wavelength of approximately 17mm, so for circular pistonic transducers, a maximum diameter of about 17mm is preferable).

The overall dimensions of the or each array of SETs in the plane of the array are very preferably chosen to be as great as or greater than the acoustic wavelength in air of the lowest frequency at which it is intended to significantly affect the polar radiation pattern of the array. Thus, if it is desired to be able to beam or steer frequencies as low as 300Hz, then the array size, in the direction at right angles to each plane in which steering or beaming is required, should be at least $c_s / 300 \doteq 1.1$metre (where c_s is the acoustic sound speed).

The invention is a fully digital steerable sonic/ audible acoustic phased array antenna system, and while the actual transducers can be driven by an analogue signal most preferably they are driven by a digital

power amplifier. A typical such digital power amplifier incorporates: a PCM signal input; a clock input (or a means of deriving a clock from the input PCM signal); an output clock, which is either internally generated, or derived from the input clock or from an additional output clock input; and an optional output level input, which may be either a digital (PCM) signal or an analogue signal (in the latter case, this analogue signal may also provide the power for the amplifier output). A characteristic of a digital power amplifier is that, before any optional analogue output filtering, its output is discrete valued and stepwise continuous, and only can only change level at intervals which match the output clock period. The discrete output values are controlled by the optional output level input, where provided. For PWM-based digital amplifiers, the output signal's average value over any integer multiple of the input sample period is representative of the input signal. For other digital amplifiers, the output signal's average value tends towards the input signal's average value over periods greater than the input sample period. Preferred forms of digital power amplifier include bipolar pulse width modulators, and one-bit binary modulators.

The use of a digital power amplifier avoids the more common requirement - found in most so-called "digital" systems - to provide a digital-to-analogue converter (DAC) and a linear power amplifier for each transducer drive channel, and therefore the power drive efficiency can be very high. Moreover, as most moving coil acoustic transducers are inherently inductive, and mechanically act quite effectively as low pass filters, it may be unnecessary to add elaborate electronic low-

pass filtering between the digital drive circuitry and the SETs.

The DPAA has one or more digital input terminals (Inputs).

In the invention the SETs are each connected to each of the inputs *via* one or more input signal distributor. At the most basic, an input signal is fed to a single Distributor, and that single Distributor has a separate output to each of the SETs (and the signal it outputs is suitably modified, as discussed hereinafter, to achieve the end desired). Alternatively, there may be a number of similar Distributors, each taking the, or part of the, input signal, or separate input signals, and then each providing a separate output to each of the SETs (and in each case the signal it outputs is suitably modified, within the Distributor, as discussed hereinafter, to achieve the end desired). In this latter case - a plurality of Distributors each feeding all the SETs - the outputs from each Distributor to any one SET have to be combined, and conveniently this is done by an adder circuit prior to any further modification the resultant feed may undergo.

The Inputs to which may be connected one or more digital signals representative of the sound or sounds to be handled by the DPAA (Input Signals). Of course, the original electrical signal defining the sound to be radiated may be in an analogue form, and therefore the system of the invention may include one or more analogue-to-digital converters (ADCs) connected each between an auxiliary analogue input terminal (Analogue Input) and one of the Inputs, thus allowing the conversion of these external analogue electrical signals to internal digital electrical signals, each with a specific (and appropriate) sample rate Fs_i. And thus,

within the DPAA, beyond the Inputs, the signals handled are time-sampled quantized digital signals representative of the sound waveform or waveforms to be reproduced by the PAA.

A digital sample-rate-converter (DSRC) is very preferably connected between each Input and the remaining internal electronic processing systems, with the output of each DSRC clocked in-phase with and at the same rate as all the other DSRCs, so that disparate external signals from the Inputs with different clock rates and/or phases can be brought together within the DPAA, synchronised, and combined meaningfully into one or more composite internal data channels. The DSRC may be omitted on one "master" channel if that input signal's clock is then used as the master clock for all the other DSRC outputs. Where several external input signals already share a common external or internal data timing clock then there may effectively be several such "master" channels.

No DSRC is required on any analogue input channel as its analogue to digital conversion process may be controlled by the internal master clock for direct synchronisation.

The DPAA of the invention incorporates a Distributor which modifies the input signal prior to feeding it to each SET in order to achieve the desired directional effect. A Distributor is a digital device, or piece of software, with one input and multiple outputs. One of the DPAA's Input Signals is fed into its input. It preferably has one output for each SET; alternatively, one output can be shared amongst a number of the SETs or the elements of a CSET. The Distributor sends generally differently modified versions of the input signal to each of its outputs. The modifications

can be either fixed, or adjustable using a control system. The modifications can comprise signal delay means (SDM), amplitude control means (ACM) and/or adjustable digital filters (ADF).

The signal-delay means (SDM) are variable digital signal time-delay elements. Here, because these are not single-frequency, or narrow frequency-band, *phase shifting* elements but true time-delays, the PAA will operate over a broad frequency band (e.g. the Audio Band). There may be means to adjust the delays between a given input terminal and each SET, and advantageously there is a separately adjustable delay means for each Input/SET combination. The minimum delay possible for a given digital signal is preferably as small or smaller than Ts, that signal's sample period; the maximum delay possible for a given digital signal should preferably be as large as or larger than Tc, the time taken for sound to cross the transducer array across its greatest lateral extent, D_{max}, where $Tc = D_{max} / c_s$ where c_s is the speed of sound in air. Most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sample period.

The amplitude control means (ACM) is conveniently implemented as digital amplitude control means for the purposes of gross beam shape modification.

An alternative way of modifying the signal uses digital filters (ADF) whose group delay and magnitude response vary in a specified way as a function of frequency (rather than just a simple time delay or level change) - simple delay elements may be used in implementing these filters to reduce the necessary computation. This approach allows control of the DPAA radiation pattern as a function of frequency. This

allows control of the radiation pattern of the DPAA to be adjusted separately in different frequency bands (which is useful because the size in wavelengths of the DPAA radiating area, and thus its directionality, is otherwise a strong function of frequency). For example, for a PAA of say 2m extent its low frequency cut-off (for directionality) is around the 150Hz region, and as the human ear has difficulty in determining directionality of sounds at such a low frequency it may be more useful not to apply "beam-steering" delays and aperture weighting at such low frequencies but instead to go for an optimized output level. Additionally, it also allows some compensation for unevenness in the radiation pattern of each SET.

The SDM delays, ACM gains and ADF coefficients can be fixed, varied in response to User input, or under automatic control. Preferably, any changes required while a channel is in use are made in many small increments so that no discontinuity is heard.

If different SETs in the array have different inherent sensitivities then it may be preferred to calibrate out such differences using an analogue method associated directly with the SETs themselves and/or their power driving circuitry, in order to minimise any loss in resolution that might result from utilising digital calibration further back in the signal processing path. This refinement is particularly useful where low-bit-number high-over-sample-rate digital coding is used prior to the points in the system where multiple input-channel-signals are brought together in combination for application to individual transducers.

Where more than one Input is provided - i.e. there are I inputs numbered 1 to I - and where there are N SETs, numbered 1 to N, it is preferable to provide a

separate and separately-adjustable delay and/or filter means D_{in}, (where *i = 1 to I, n = 1* to *N,* between each of the *I* inputs and each of the *N* SETs. For each SET there are thus *I* delayed or filtered digital signals, one from each of the Inputs *via* the separate Distributor, to be combined before application to the SET. As noted above, this combination of digital signals is conveniently done by digital algebraic addition of the *I* separate delayed signals - i.e. the signal to each SET is a linear combination of separately modified signals from each of the *I* Inputs. It is because of this requirement to perform digital addition of signals originating from more than one Input that the DSRCs (see above) are desirable, to synchronize these external signals, as it is generally not meaningful to perform digital addition on two or more digital signals with different clock rates and or phases.

With the basic digital sonic PAA as described above, the following modes of operation are possible:-

By setting all the signal delays (between an Input/ Distributor and each SET) to the same value, e.g. 0 (in the case of plane Surfaces), or to values that are a function of the shape of the Surface (in the case of curved Surfaces), the array can be made to produce a roughly parallel "beam" of sound representative of the signal at that Input, at right angles to the tangents to the centre of the Surface of the PAA. The radiation in the direction of this beam is then significantly more intense than in other directions, though in general there will be "side lobes" too; the assumption is that the PAA array physical extent is one or several wavelengths at the sound frequencies of interest.

As discussed hereinafter, the side lobes can generally be attenuated or moved if necessary by adjustment of the ACMs or ADFs.

By varying the signal delay between an Input/ Distributor and each SET such that the delay increases systematically amongst the SETs in some *chosen direction* across the Surface, a similar roughly parallel "beam" of sound will again be produced, again at right angles to that tangent to the centre of the Surface of the PAA that is at right angles to the *chosen direction*, and at an angle to the Surface of the PAA other than a right-angle, the angle being dependent on the amount of systematic delay increase in the orthogonal direction. For very small delays (d_i <<Tc, for all i), the beam direction will be very nearly orthogonal to the Surface; for larger delays (max(d_i) ~ Tc) the beam can be steered to be nearly tangential to the Surface.

By reducing the amplitudes of the signals presented by a Distributor to the SETs closer to the edges of the array (relative to the amplitudes presented to the SETs closer to the middle of the array), the level of the side lobes (due to the finite array size) in the radiation pattern may be reduced.

If the signal delay between an Input and each SET is chosen such that the sum of that delay plus the sound travel time from that SET to a chosen point in space in front of the PAA are for all the SETs all the same value - i.e. so that Input signals converted to sound by the PAA arrive at the chosen point as in-phase sounds - then the PAA may be caused to *focus* sound at that point. In other words, the sound intensity at and around that point (in a region of dimensions roughly equal to a wavelength of each of the spectral components of the sound) is considerably higher than at other points

nearby. The position of the *focal point* may be varied widely almost anywhere in front of the PAA by suitably choosing the set of delays as previously described.

If the signal delay between an Input and each SET is chosen to equal the sound travel time to that SET from a point in space behind the array, then, for listeners in the near field in front of the array, the sound will appear to come from that point.

Where several Inputs/Distributors are provided, they may each be separately and distinctly *focused* in the manner described above, because of the linear combination of signals in the PAA as previously described, giving rise to controllably and potentially widely separated regions where distinct sound fields (representative of the signals at the different Inputs) may be distinguished, remote from the PAA proper.

Where the DDPAA is operated in *anechoic conditions* in the manner previously described - i.e. with the sound signals representative of distinct Input signals *focused* in distinct and separated *focal regions* -- an observer needs to move close to the separate focal regions easily to perceive the separate sound fields, and it is difficult otherwise to locate them in space.

If an acoustic reflecting surface, or alternatively an acoustically-resonant body which re-radiates absorbed incident sound energy, is placed in such a focal region, it re-radiates the focused sound, and so effectively becomes a new sound source, remote from the DPAA, and located at the focal region. If a plane reflector is present then the reflected sound is predominantly directed in a specific direction; if a diffuse reflector is present then the sound is re-radiated more or less in all directions away from the focal region on the same side of the reflector as the focused sound is incident

from the DPAA. Thus, if a number of distinct sound signals representative of distinct Input signals are focused to distinct focal regions by the DPAA in the manner described, and within each focal region is placed such a reflector or resonator so as to redirect the sound from each focal region, then a true multiple separated-source sound radiator system may be constructed using a single DPAA of the design described herein.

Where the DPAA is operated in the manner previously described with multiple separated focused beams - i.e. with sound signals representative of distinct Input signals *focused* in distinct and separated regions - in *non-anechoic* conditions (such as in a normal room environment) wherein there are multiple hard and/or predominantly sound reflecting boundary surfaces, and in particular where those focused regions are directed at one or more of the reflecting boundary surfaces, then using only his normal directional sound perceptions an observer is easily able to perceive the separate sound fields, and simultaneously locate each of them in space *at* their respective separate focal regions, due to the reflected sounds (from the boundaries) reaching the observer *from* those regions.

It is important to emphasise that in such a case the observer perceives *real* separated sound fields which in no way rely on the DPAA introducing artificial psycho-acoustic elements into the sound signals. Thus, the position of the observer is relatively unimportant for true sound location, so long as he is sufficiently far from the near-field radiation of the DPAA. In this manner, multi-channel "surround-sound" can be achieved with only one physical loudspeaker (the DPAA), making use of the natural boundaries found in most real environments.

Where similar effects are to be produced in an environment lacking appropriate natural reflecting boundaries, similar separated multi-source sound fields can be achieved by the suitable placement of artificial reflecting or resonating surfaces where it is desired that a sound source should seem to originate, and then directing focused beams at those surfaces. For example, in a large concert hall or outside environment optically-transparent plastic or glass panels could be placed and used as sound reflectors with little visual impact. Where wide dispersion of the sound from those regions is desired, a sound scattering reflector or broadband resonator could be introduced instead (this would be more difficult but not impossible to make optically transparent).

In certain circumstances, especially in concert hall and arena settings, it is also possible to make use of the fact that the DPAA is front-back symmetrical in its radiation pattern, when beams with real focal points are formed, in the case where the array of transducers is made with an open back (i.e. no sound-opaque cabinet placed around the rear of the transducers). For example, in the instance described above where sound reflecting or scattering surfaces are placed near such real foci at the "front" of the DPAA, additional such reflecting or scattering surfaces may advantageously be positioned at the mirror image real focal points behind the DPAA to further direct the sound in the desired manner. In particular, is a DPAA is positioned with its side facing the target audience area, and an off-axis beam from the front of the array is steered to a particular section of the audience, say at the left of the auditorium, then its mirror-image focused beam (in antiphase) from the rear of the DPAA will be directed to a well-separated section of the same audience at the

right of the auditorium. In this manner useful acoustic power may be derived from both the front and rear radiation fields of the transducers;

The *basic* DPAA system, and some possible uses, has now been described. However, in a further refinement of the DPAA system described above, giving rise to yet further uses and advantages, the following additional features may be added:-

The input digital signals are very preferably passed through an oversampling-noise-shaping-quantizer (ONSQ) which reduces their bit-width and increases their sample-rate whilst keeping their signal to noise ratio (SNR) in the acoustic band largely unchanged. The principle reason for doing this is to allow the digital transducer drive-circuitry ("digital amplifiers") to operate with feasible clock rates. For example, if the drives are implemented as digital PWM, then if the signal bit-width to the PWM circuit is *b* bits, and its sample rate *s* samples per second, then the PWM clock-rate *p* needs to be $p = 2^b s$ Hz - e.g. for b = 16, and s = 44KHz, then p = 2.88GHz, which quite impractical at the present level of technology. If, however, the input signal were to be oversampled 4 times and the bit width reduced to 8 bits, then $p = 2^8 \times 4 \times 44$KHz = 45MHz, which is easily achievable with standard logic or FPGA circuitry. In general, use of an ONSQ increases the signal bit rate. In the example given the original bit rate $R_o = 16 \times 44000$ = 704Kbits/sec, whilst the oversampled bit rate is
$R_q = 8 \times 44000 \times 4$ = 1.408Mbits/sec, which is twice as high). If the ONSQ is connected between an Input and the inputs to the digital delay generators (DDG), then the DDG will in general require more storage capacity to

accommodate the higher bit rate; if, however, the DDGs operate at the Input bit-width and sample rate (thus requiring the minimum storage capacity in the DDGs), and instead an ONSQ is connected between each DDG output and SET digital driver, then one ONSQ is required for every SET, which increases the complexity of the DPAA, where the number of SETs is large. There are two additional trade-offs in the latter case:

> the DDG circuitry can operate at a lower clock rate, subject to the requirement for sufficiently fine control of the signal delays; and
> with an array of separate ONSQs the quantization-noise from each can be designed to be uncorrelated with the noise from all the rest, so that at the output of the DPAA the quantization-noise components will add in an uncorrelated fashion and so each doubling of the number of SETs will lead to an increase of only 3dB instead of 6dB to the total quantization-noise power;

and these considerations may make post-DDG ONSQs (or two stages of OSNQ - one pre-DDG and one post-DDG) the more attractive implementation strategy.

The input digital signal(s) are advantageously passed through one or more digital pre-compensators to correct for the linear and/or non-linear response characteristics of the SETs. In the case of a DPAA with multiple Inputs/Distributors, it is essential that, if non-linear compensation is to be carried out, it be performed on the digital signals *after* the separate channels have been combined in the digital adders which occur after the DDGs too; this results in the requirement for a separate non-linear compensator (NLC) for each and every SET. However, in the case of linear-compensation, or where there is only one Input/

Distributor, the compensator(s) can be placed directly in the digital signal stream after the Input(s), and at most one compensator per Input is required. Such linear compensators are usefully implemented as filters which correct the SETs for amplitude and phase response across a wide frequency range; such non-linear compensators correct for the imperfect (non-linear) behaviour of the SET motor and suspension components which are generally highly non-linear where considerable excursion of the SET moving-component is required.

One or more microphones may be provided that are able to sense the acoustic emission from the DPAA, and which are connected to the DPAA control electronics either by wired or wireless means. The DPAA can then incorporate a subsystem arranged to be able to compute the location of the microphone(s) relative to the DPAA SET array by measuring the propagation times of signals from three or more (and in general from all of the) SETs to the microphone and triangulating, thus allowing the possibility of tracking the microphone movements during use of the DPAA without interfering with the listener's perception of the programme material sound. Where the DPAA SET array is open-backed - i.e. it radiates from both sides of the transducers, in a dipole like manner - the potential ambiguity of microphone position, in front of or behind the DPAA, may be resolved by examination of the phase of the received signals (especially at the lower frequencies). The speed of sound, which changes with air temperature during the course of a performance, affecting the acoustics of the venue and the performance of the speaker system, can be determined in the same process by using an additional triangulation point. The microphone locating may either be done using a specific test pattern (e.g. a pseudo-random noise sequence or sequence of short pulses to each of the SETs in turn,

where the pulse length t_p is as short or shorter than the spatial resolution r_s required, in the sense that $t_p \le r_s / c_s$) or by introducing low level test signals (which may be designed to be inaudible) with the programme material being broadcast by the DPAA, and then detecting these by cross-correlation.

The DPAA system may be used with a remote-control handset (Handset) that communicates with the DPAA electronics (*via* wires, or radio or infra-red or some other wireless technology) over a distance (ideally from anywhere in the listening area of the DPAA), and provides manual control over all the major functions of the DPAA. Such a control system would be most useful to provide the following functions:

selection of which Input(s) are to be connected to which Distributor, which might also be termed a "Channel";

the focus position and/or beam shape of each Channel;

the individual volume-level settings for each Input;

set-up using Handset microphone; and

the various modes of optimization to be performed.

A control system may be added to the DPAA that optimises (in some desired sense) the sound field at one or more specified locations, by altering the DDGs and/or the filter coefficients of the ADFs. If the previously described microphones are available, then this optimization can occur either at set-up time - for instance during pre-performance use of the DPAA) - or during actual use. In the latter case, one or more of the microphones may be embedded in the Handset used otherwise to control the DPAA, and in this case the

control system may be designed actively to track the microphone in real-time and so continuously to optimize the sound at the position of the handset, and thus at the presumed position of at least one of the listeners. By building into the control system a model (most likely a software model) of the DPAA and its acoustic characteristics, plus optionally a model of the environment in which it is currently situated (i.e. where it is in use, e.g. a listening room), the control system may use this model to estimate automatically the required adjustments to the DPAA parameters to optimize the sound at any user-specified positions to reduce any troublesome side lobes.

The control system just described can additionally be made to adjust the sound level at one or more specific locations - e.g. positions where live performance microphones are situated, which are connected to the DPAA, or positions where there are known to be undesired reflecting surfaces - to be minimised, creating "dead-zones". In this way unwanted mic/DPAA feedback can be avoided, as can unwanted room reverberations.

By using buried test-signals - that is, additional signals generated in the DPAA electronics which are designed to be largely imperceptible to the audience, and typified by low level pseudo random noise sequences, which are superimposed on the programme signals - one or more of the live performance microphones can be spatially tracked (by suitable processing of the pattern of delays between said microphones and then DPAA transducers). This microphone spatial information may in turn be used for purposes such as positioning the "dead-zones" wherever the microphones are moved to (note

that the buried test-signals will of necessity be of non-zero amplitude at the microphone positions).

In outdoor performances, wind has a significant impact on the performance of loudspeaker systems. The direction of propagation of sound is affected by winds. In particular, wind blowing across an audience, at perpendicular to the desired direction of propagation of the sound, can cause much of the sound power to be delivered outside the venue, with insufficient coverage within. With a DPAA system, the propagation of the microphone location finding signals are affected in the same manner by crosswinds. Hence, if a microphone was positioned in the middle of the audience area, but a crosswind was blowing from the North, it would appear to the location finding system than the microphone is North of the audience area. If the radiation pattern of the array was then adjusted to optimize coverage around the apparent microphone location, this would compensate for the wind, and give optimum coverage in the actual audience area. The DPAA control system can make these adjustments automatically during the course of a performance. To ensure stability of the control system, only slow changes must be made. The robustness of the system can be improved by using multiple microphones at known locations throughout the audience area.

Where it is desired to position an apparent source of sound remote from the DPAA as previously described (by there focusing a beam of sound energy onto a suitable reflecting surface), the use of the microphones previously described allows a simple way to set up this situation. One of the microphones is temporarily positioned near the surface which is to become the remote sound source, and the position of the microphone is accurately determined by the DPAA sub-system already

described. The control system then computes the optimum array parameters to locate a focused beam (connected to one or more of the user-selected Inputs) at the position of the microphone. Thereafter, the microphone may be removed. The separate remote sound source will then emanate from the surface at the chosen location.

Where a user wishes to specify the radiation pattern, the use of ADFs allows a constrained optimization procedure many degrees of freedom. A user would specify targets, typically areas of the venue in which coverage should be as even as possible, or should vary systematically with distance, other regions in which coverage should be minimised, possibly at particular frequencies, and further regions in which coverage does not matter. The regions can be specified by the use of microphones or another positioning system, by manual user input, or through the use of data sets from architectural or acoustic modelling systems. The targets can be ranked by priority. The optimization procedure can be carried out either by within the DPAA itself, in which case it could be made adaptive in response to wind variations, as described above, or as a separate step using an external computer.

If multiple sources are used simultaneously in a DPAA, to avoid clipping or distortion, it is important to ensure that none of the summed signals presented to the SETs exceed the maximum excursion of the SET pistons or the full-scale digital level (FSDL) of the summing units, digital amplifiers, ONSQs or linear or non-linear compensators. This can be achieved straightforwardly by either scaling down or peak limiting each of the I input signals so that no peak can exceed 1/Ith of the full scale level. This approach caters for the worst case, where all input signals peak at the FSDL together, but

severely limits the output power available to a single input. In most applications this is unlikely to occur except during occasional brief transients (such as explosions in a movie soundtrack). Better use can therefore be made of the dynamic range of the digital system if higher levels are used and overload avoided by peak limiting only during such simultaneous peaks.

A digital peak limiter is a system which scales down an input digital audio signal as necessary to prevent the output signal from exceeding a specified maximum level. It derives a control signal from the input signal, which may be subsampled to reduce the required computation. The control signal is smoothed to prevent discontinuities in the output signal. The rate at which the gain is decreased before a peak (the attack time constant) and returned to normal afterwards (the release time constant) are chosen to minimise the audible effects of the limiter. They can be factory-preset, under the control of the user, or automatically adjusted according to the characteristics of the input signal. If a small amount of latency can be tolerated, then the control signal can "look ahead" (by delaying the input signal but not the control signal), so that the attack phase of the limiting action can anticipate a sudden peak.

Since each SET receives sums of the input signals with different relative delays, it is not sufficient simply to derive the control signal for a peak limiter from a sum of the input signals, as peaks which do not coincide in one sum may do so in the delayed sums presented to one or more SETs. If independent peak limiters are used on each summed signal then, when some SETs are limited and others are not, the radiation pattern of the array will be affected.

This effect can be avoided by linking the limiters so that they all apply the same amount of gain reduction. This, however, is complex to implement with when N is large, as it generally will be, and does not prevent overload at the summing point.

An alternative approach is the Multichannel Multiphase Limiter (MML). This acts on the input signals. It finds the peak level of each input signal in a time window spanning the range of delays currently implemented by the SDMs, then sums these I peak levels to produce its control signal. If the control signal does not exceed the FSDL, then none of the delayed sums presented to individual SETs can, so no limiting action is required. If it does, then the input signals should be limited to bring the level down to the FSDL. The attack and release time constants and the amount of lookahead can be either under the control of the user or factory-preset according to application.

If used in conjunction with ONSQ stages, the MML can act either before or after the oversampler.

Lower latency can be achieved by deriving the control signal from the input signals before oversampling, then applying the limiting action to the oversampled signals; a lower order, lower group delay anti-imaging filter can be used for the control signal, as it has limited bandwidth.

There may also be:

means to interconnect two or more such DPAAs in order to coordinate their radiation patterns, their focusing and their optimization procedures;

means to store and recall sets of delays (for the DDGs) and filter coefficients (for the ADFs);

means to detect failed SETS in the array, either by built-in monitoring circuits, or by use of the

microphone(s) and test signals previously described - when a failed SET is detected, the operator may be alerted, and additionally delays and filter coefficients in the remainder of the functional DPAA adjusted to minimise the effects of the failure;

means to adjust the radiation pattern and focusing points of signals related to each Input, in response to the value of the programme digital signals at those Inputs - such an approach may be used to exaggerate stereo signals and surround-sound effects, by moving the focusing point of those signals momentarily outwards when there is a loud sound to be reproduced from that Input only; and

means to project one or more steerable beams of visible light from the DPAA:

if one such light beam is provided it may be steered to point in the same general direction as one of the DPAA sound beams (Channels) and thus indicate to the operator the general set up of that Channel;

if two light beams are provided, then they may be steered automatically by the DPAA electronics such that they intersect in space at or near the centre of the focal region of a Channel, again providing a great deal of useful set-up feedback information to the operator;

it is useful to make the colours of the two beams different, and different primaries may be best, e.g. red and green, so that in the overlap region a third colour is perceived;

means to select which Channel settings control the positions of the light beams should also be

provided and these may all be controlled from the handset;

where more than two light beams are provided, the focal regions of multiple Channels may be high-lighted simultaneously by the intersection locations in space of pairs of the steerable light beams;

small laser beams, particularly solid-state diode lasers, provide a useful source of collimated light;

steering is easily achieved through small steerable mirrors driven by galvos or motors, or alternatively by a WHERM mechanism as described in the Specification of British Patent Application No: 00/03,136.9 (P1591Sub).

Practical applications of this invention's DPAA technology include the following:

for home entertainment, the ability to project multiple real sources of sound to different positions in a listening room, allow the reproduction of multi-channel surround sound without the clutter, complexity and wiring problems of multiple separated wired loudspeakers;

for public address and concert sound systems, the ability to tailor the radiation pattern of the DPAA in three dimensions, and with multiple simultaneous beams allows:

much faster set-up as the physical orientation of the DPAA is not very critical and need not be repeatedly adjusted;

smaller loudspeaker inventory as one type of speaker (a DPAA) can achieve a wide variety of

radiation patterns which would typically each require dedicated speakers with appropriate horns;

better intelligibility, as it is possible to reduce the sound energy reaching reflective surfaces, hence reducing dominant echoes, simply by the adjustment of filter and delay coefficients; and

better control of unwanted acoustic feedback as the DPAA radiation pattern can be designed to reduce the energy reaching live microphones connected to the DPAA input;

for crowd-control and military activities, the ability to generate a very intense sound field in a distant region, which field is easily and quickly repositionable, by focusing and steering of the DPAA beams (without having physically to move bulky loudspeakers and/or horns) and which is easily directed onto the target by means of tracking light sources, and provides a powerful acoustic weapon which is nonetheless non-invasive; if a large array is used, or a group of coordinated separate DPAA panels possibly widely spaced, then the sound field can be made much more intense in the focal region than near the DPAA SETs (even at the lower end of the Audio Band if the overall array dimensions are sufficiently large).

Embodiments of the several aspects of the invention are now described, though by way of illustration only, with reference to the accompanying diagrammatic Drawings in which:

Figure 1	is a schematic representation of a simple single-input DPAA;
Figures 2A & B	are front and orthogonal views of a multiple-Surface array;
Figures 3A & B	are front view of an array of CSETs and an array of multiple types of SET;
Figures 4A & B	are front views of rectangular and hexagonal lattice arrays of SETs.
Figure 5	is a block diagram of a DPAA with multiple inputs;
Figure 6	is a block diagram of the input stages of a DPAA with its own master clock;
Figure 7	is a block diagram of the input stages of a DPAA which recovers an external clock;
Figure 8	is a block diagram of a Distributor;
Figure 9	is a plan view of the use of a DPAA to convey multiple channels of sound to an audience;
Figure 10	is a plan view of the use of a DPAA in end-fire mode, utilising the rear radiation of the array;

Figure 11	is a block diagram of possible linear and digital amplifier configurations;
Figure 12	is a block diagram showing the points at which ONSQ stages can be incorporated into a DPAA;
Figure 13	is a block diagram showing where linear and non-linear compensation can be incorporated into a single-input DPAA;
Figure 14	is a block diagram showing where linear and non-linear compensation can be incorporated into a more general DPAA;
Figure 15	is an illustration of the use of test signals and triangulation to determine the position of a microphone;
Figure 16	is a block diagram showing how test signal generation and analysis can be incorporated into a DPAA;
Figure 17	is a block diagram showing two alternative means of inserting test signals into the feed to a SET;
Figure 18	is a block diagram of an MML;
Figure 19	is a block diagram illustrating the interconnection of several DPAAs with common control and input stages; and

<u>Figure 20</u> is a plan view illustrating the use of light beams to indicate focal points.

The Figures generally only show audio signal paths; clock and control connections are omitted for clarity unless necessary to convey the idea. Moreover, only small numbers of SETs, Channels, and their associated circuitry are shown, as diagrams become cluttered and hard to interpret if the realistically large numbers of elements are included.

The block diagram of Figure 1 depicts the simplest DPAA. An input signal (101) feeds a Distributor (102) whose many outputs each connect through amplifiers (103) to SETs (104) which are physically arranged to form a two-dimensional array (105). The distributor modifies the signal sent to each SET to produce the desired radiation pattern. There may be additional processing steps before and after the distributor, which are illustrated in turn later. Details of the amplifier section are shown in Figure 11.

Figure 2 shows SETs 104 arranged to form a front Surface (201) and a second Surface (202) such that the SETs on the rear Surface radiate through the gaps between SETs in the front Surface.

Figure 3 shows CSETs (301) arranged to make an array (302), and two different types of SET (303,304) combined to make an array (305).

Figure 4 shows two possible arrangements of SETs 104 forming a rectangular array (401) and a hex array (402).

Figure 5 shows a DPAA with two input signals (501,502) and three Distributors (503-505). Distributor 503 treats the signal 501, whereas both 504 and 505 treat the input signal 502. The outputs from each Distributor for each SET are summed by adders (506), and pass through amplifiers 103 to the SETs 104. Details of the input section are shown in Figures 6 and 7.

Figure 6 shows a possible arrangement of input circuitry with, for illustrative purposes, three digital inputs (601) and one analogue input (602). Digital receiver and analogue buffering circuitry has been omitted for clarity. There is an internal master clock source (603), which is applied to DSRCs (604) on each of the digital inputs and the ADC (605) on the analogue input. Most current digital audio transmission formats (e.g. S/PDIF, AES/EBU), DSRCs and ADCs treat (stereo) pairs of channels together. It may therefore be most convenient to handle Input Channels in pairs.

Figure 7 shows an arrangement in which there are two digital inputs (701) which are known to be synchronous and from which the master clock is derived using a PLL or other clock recovery means (702). This situation would arise, for example, where several channels are supplied from an external surround sound decoder. This clock is then applied to the DSRCs 604 on the remaining inputs 601.

Figure 8 shows the components of a Distributor. It has a single input signal (801) coming from the input circuitry and multiple outputs (802), one for each SET or group of SETs. The path from the input to each of the outputs contains a SDM (803) and/or an ADF (804) and/or an ACM (805). If the modifications made in each signal path are similar, the Distributor can be

implemented more efficiently by including global SDM, ADF and/or ACM stages (806-808) before splitting the signal. The parameters of each of the parts of each Distributor can be varied under User or automatic control. The control connections required for this are not shown.

Figure 9 illustrates the use of a single DPAA and multiple reflecting or resonating surfaces (902) to present multiple sources to listeners (903). As it does not rely on psychoacoustic cues, the surround sound effect is audible throughout the listening area.

Figure 10 illustrates the use of an open-backed DPAA (1001) to convey a signal to left and right sections of an audience (1002,1003), exploiting the rear radiation. The different parts of the audience receive signals with opposite polarity.

Figure 11 shows possible power amplifier configurations. In one option, the input digital signal (1101), possibly from a Distributor or adder, passes through a DAC (1102) and a linear power amplifier (1103) with an optional gain/volume control input (1104). The output feeds a SET or group of SETs (1105). In a preferred configuration, this time illustrated for two SET feeds, the inputs (1106) directly feed digital amplifiers (1107) with optional global volume control input (1108). The global volume control inputs can conveniently also serve as the power supply to the output drive circuitry. The discrete-valued digital amplifier outputs optionally pass through analogue low-pass filters (1109) before reaching the SETs 1105.

Figure 12 shows that ONSQ stages can be incorporated in to the DPAA either before the Distributors, as (1201), or after the adders, as (1202),

or in both positions. Like the other block diagrams, this shows only one elaboration of the DPAA architecture. If several elaborations are to be used at once, the extra processing steps can be inserted in any order.

Figure 13 shows the incorporation of linear compensation (1301) and/or non-linear compensation (1302) into a single-Distributor DPAA. Non-linear compensation can only be used in this position if the Distributor applies only pure delay, not filtering or amplitude changes.

Figure 14 shows the arrangement for linear and/or non-linear compensation in a multi-Distributor DPAA. The linear compensation 1301 can again be applied at the input stage, but now each output must be separately non-linearly compensated 1302. This arrangement also allows non-linear compensation where the Distributor filters or changes the amplitude of the signal.

Figure 15 illustrates a possible configuration for the use of a microphone to specify locations in the listening area. The microphone (1501) is connected an analogue or digital input (1504) of the DPAA 104 *via* a radio transmitter (1502) and receiver (1503). A wired connection could instead be used if more convenient. Most of the SETs 104 are used for normal operation or are silent. A small number of SETs (1505) emit test signals, either added to or instead of the usual programme signal. The path lengths (1506) between the test SETs and the microphone are deduced by comparison of the test signals and microphone signal, and used to deduce the location of the microphone by triangulation. Where the signal to noise ratio of the received test signals is poor, the response can be integrated over several seconds.

Figure 16 shows a block diagram of the incorporation of test signal generation and analysis into a DPAA. Test signals are both generated and analysed in block (1601). It has as inputs the normal input channels 101, in order to design test signals which are imperceptible due to masking by the desired audio signal, and microphone inputs 1504. The usual input circuitry, such as DSRCs and/or ADCs have been omitted for clarity. The test signals are emitted either by dedicated SETs (1603) or shared SETs 1505. In the latter case the test signal is incorporated into the signal feeding each SET in a test signal insertion step (1602).

Figure 17 shows two possible test signal insertion steps. The programme input signals (1701) come from a Distributor or adder. The test signals (1702) come from block 1601 in Figure 16. The output signals (1703) go to ONSQs, non-linear compensators, or directly to amplifier stages. In insertion step (1704), the test signal is added to the programme signal. In insertion step (1705), the test signal replaces the programme signal. Control signals are omitted.

Figure 18 illustrates a two-channel implementation of the MML. The input signals (1801) come from the input circuitry or the linear or non-linear compensators. The output signals (1811) go to the Distributors. Each delay unit (1802) stores a number of samples of its input signal and outputs the maximum absolute value contained in its buffer as (1803). The length of the buffer can be changed to track the range of delays implemented in the Distributors by control signals which are not illustrated. The adder (1804) sums these maximum values from each channel. Its output is converted by the response shaper (1805) into a more

smoothly varying gain control signal with specified attack and release rates. Before being sent on the Distributors as 1811, in stage (1810) the input signals are all multiplied by the gain control signal. Delays (1809) may be incorporated into the channel signal paths in order to allow gain changes to anticipate peaks.

If oversampling is to be incorporated, it can be placed within the MML, with upsampling stages (1806) followed by anti-image filters (1807,1808). High quality anti-image filters can have considerable group delay in the passband. Using a filter design with less group delay for 1808 can allow the delays 1809 to be reduced or eliminated.

If the Distributors incorporate global ADFs (807), the MML is most usefully incorporated after them in the signal path, splitting the Distributors into separate global and per-SET stages.

Figure 19 illustrates the interconnection of three DPAAs (1901). In this case, the inputs (1902), input circuitry (1903) and control systems (1904) are shared by all three DPAAs. The input circuitry and control system could either be separately housed or incorporated into one of the DPAAs, with the others acting as slaves. Alternatively, the three DPAAs could be identical, with the redundant circuitry in the slave DPAAs merely inactive.

Figure 20 illustrates the use of steerable light beams (2003,2004) emitted from projectors (2001,2002) on a DPAA to show the point of focus (2005). If projector 2001 emits red light and 2002 green light, then yellow light will be seen at the point of focus.

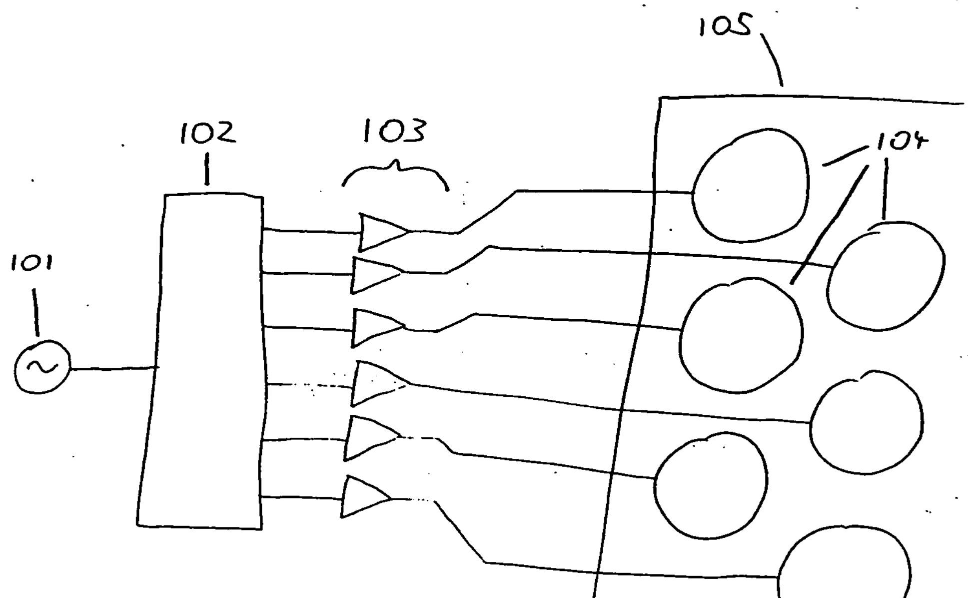

Fig 1

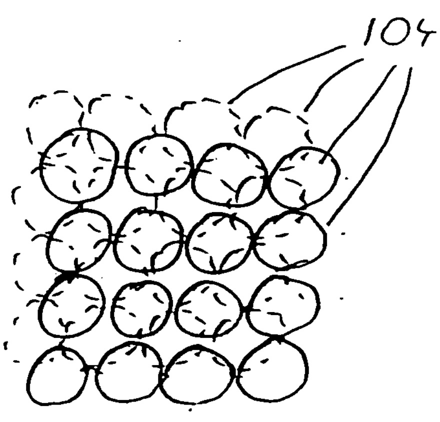

Fig 2A

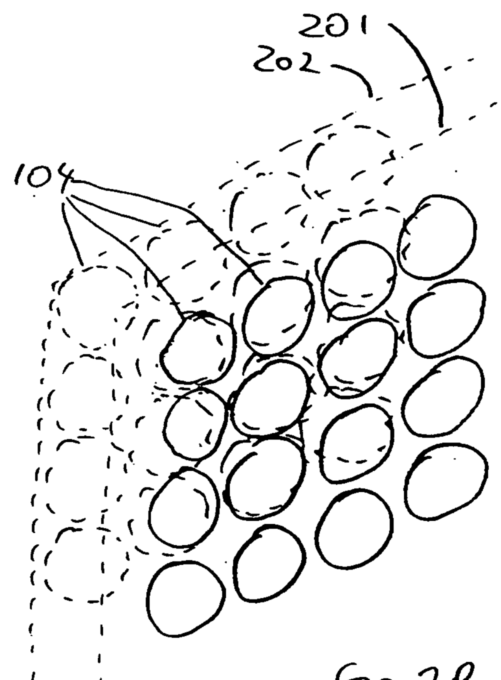

Fig 2B

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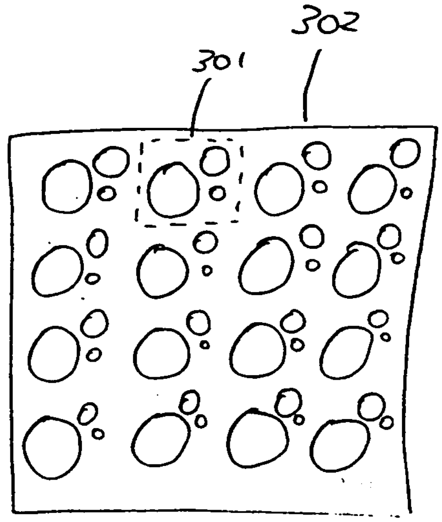

Fig 3A

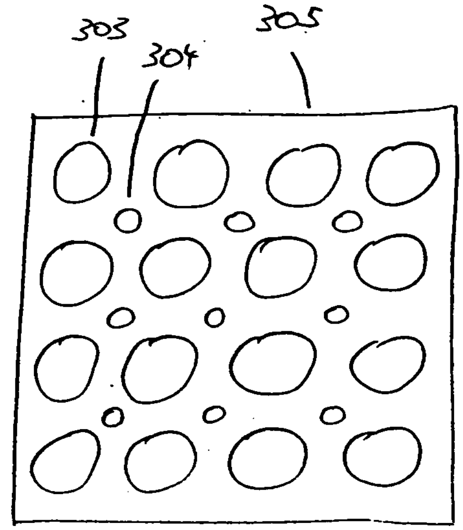

Fig 3B

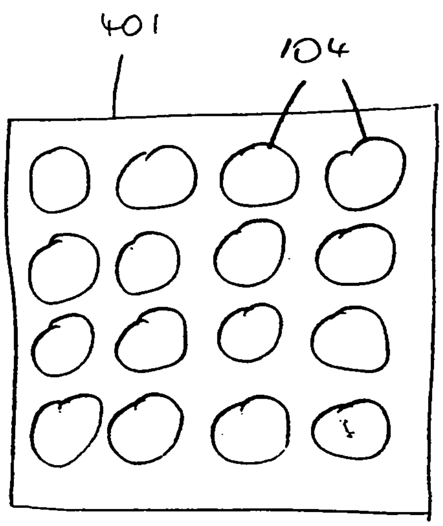

Fig 4A

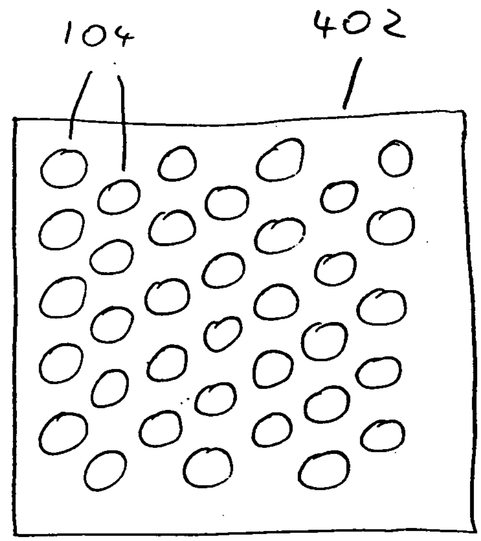

Fig 4B

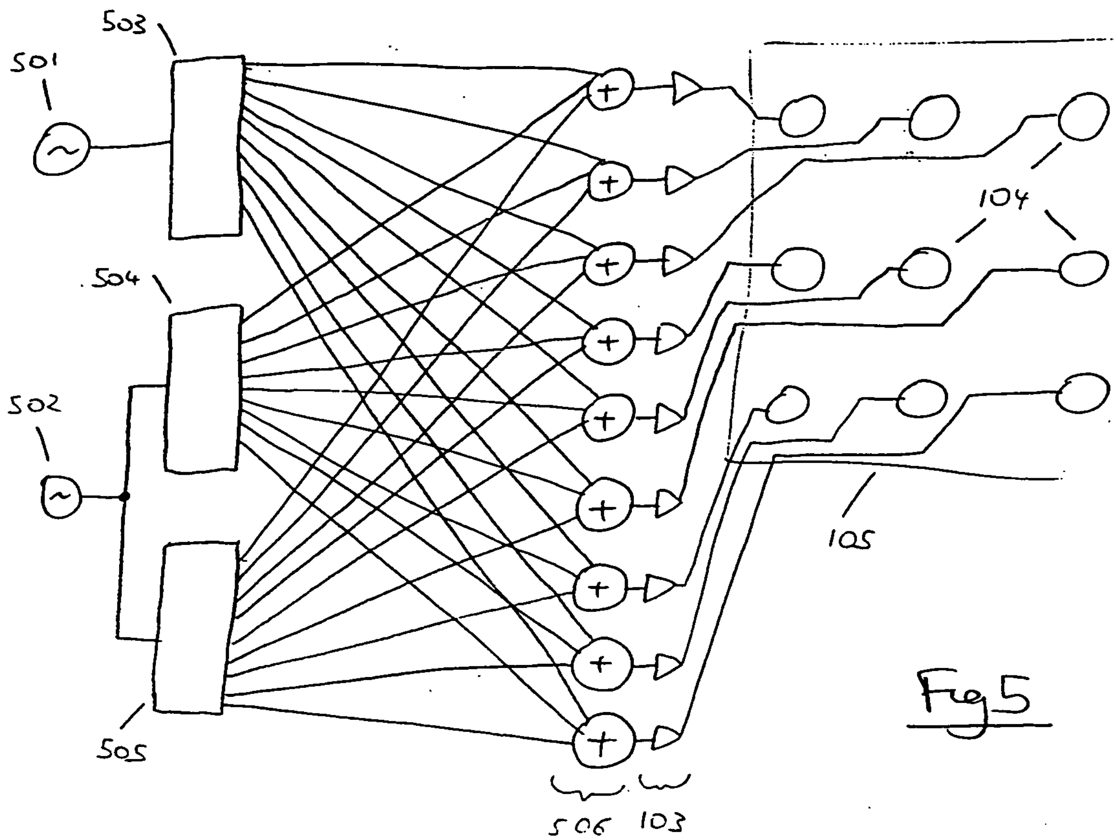

Fig 5

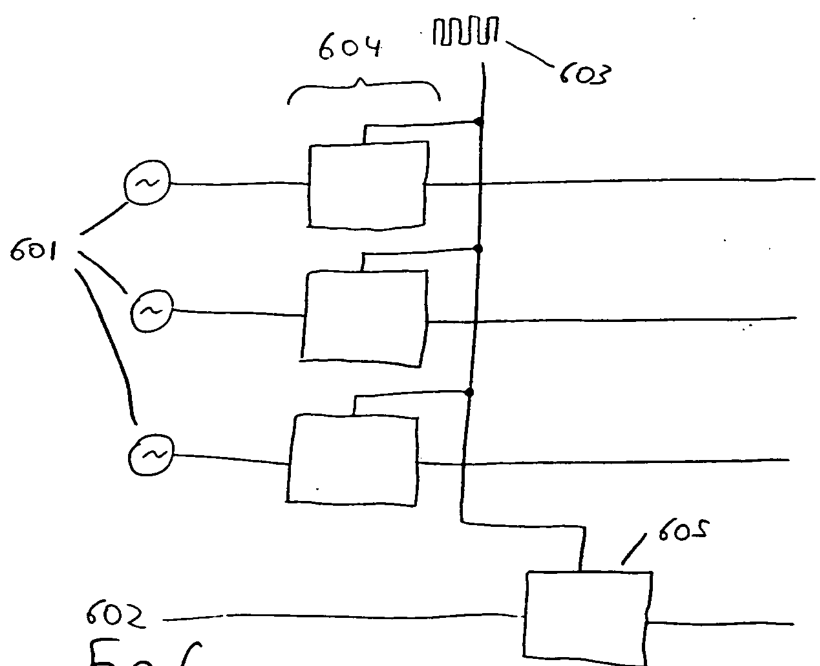

Fig 6

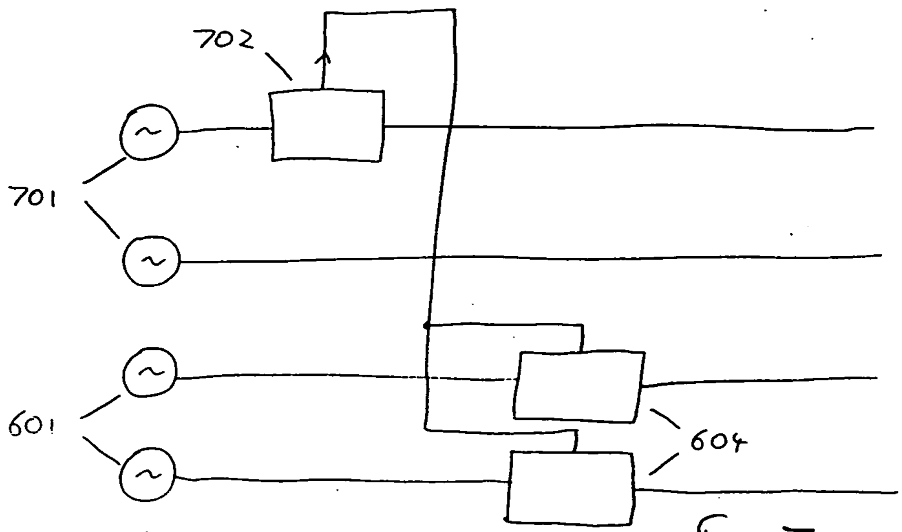

Fig 7

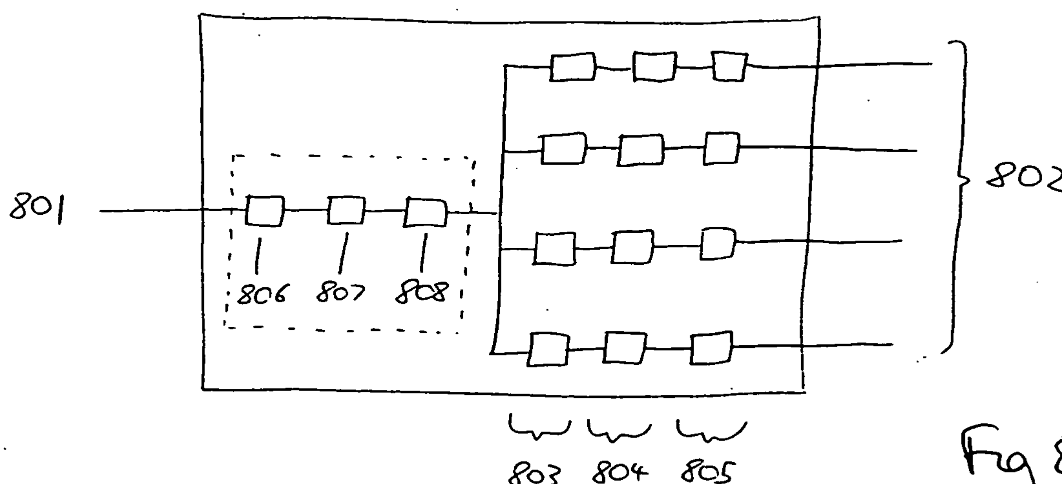

Fig 8

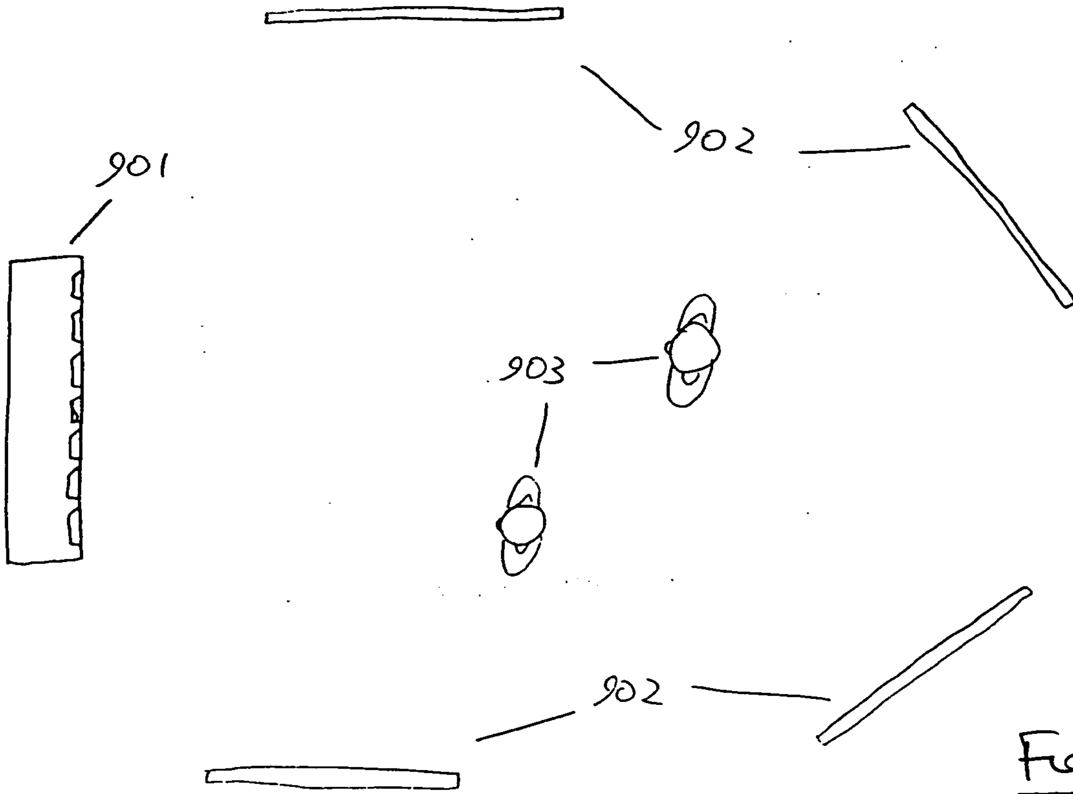

Fig 9

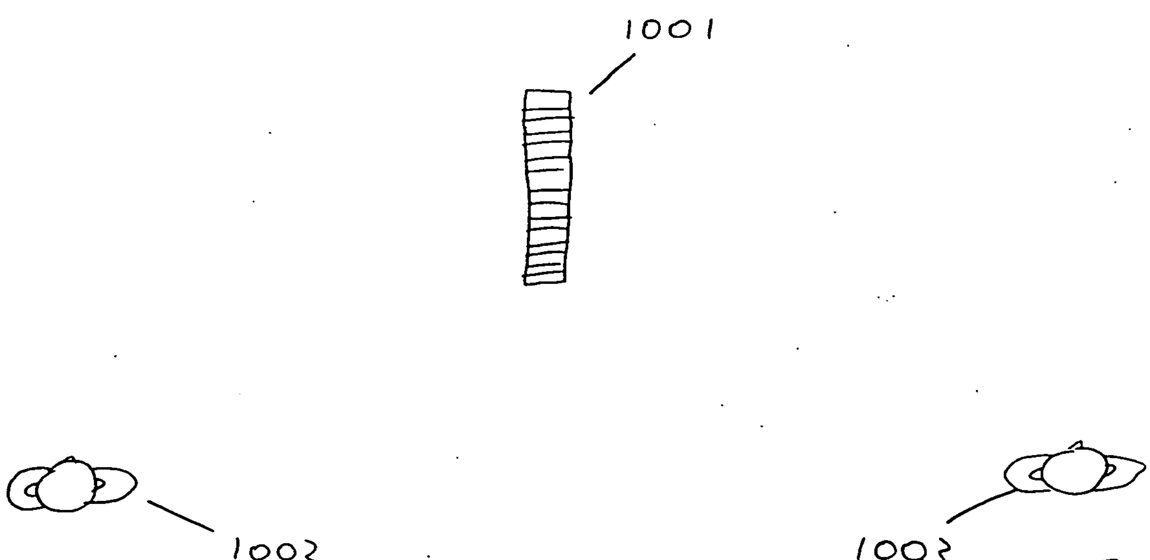

Fig 10

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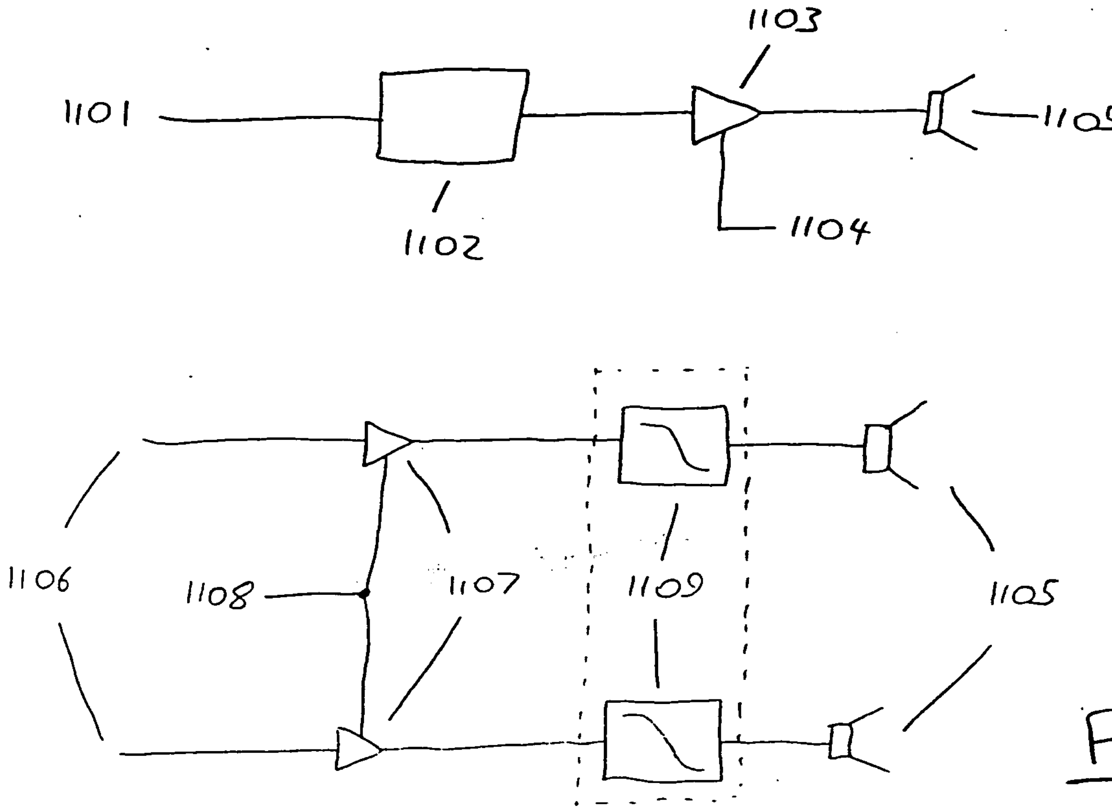
1101
1102
1103
1104
1105
1106
1108
1107
1109
1105

Fig 11

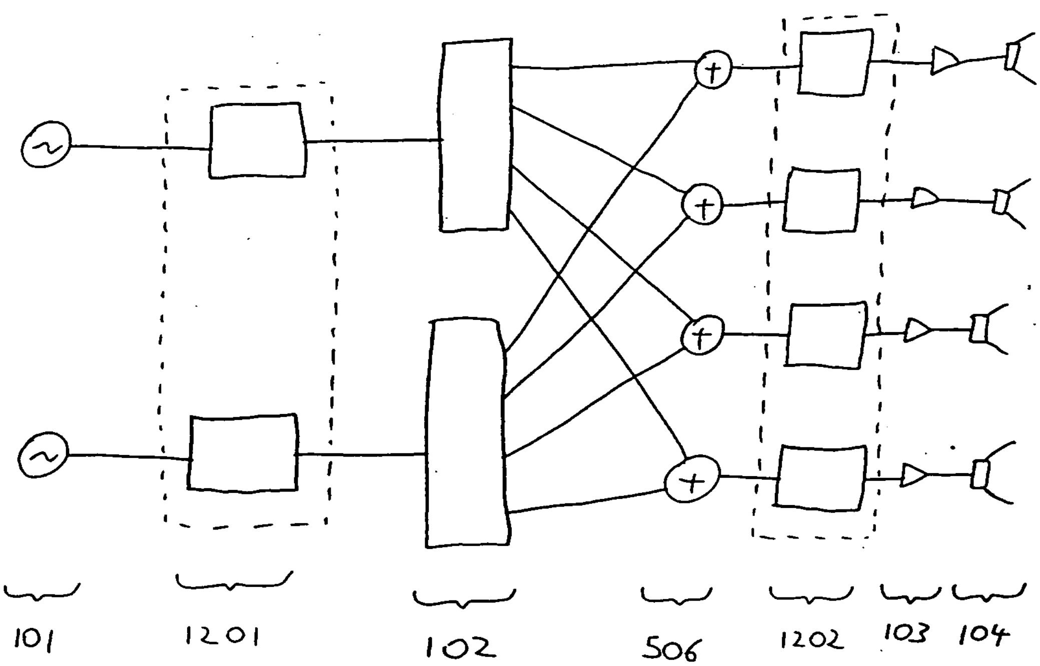
101
1201
102
506
1202
103
104

Fig 12

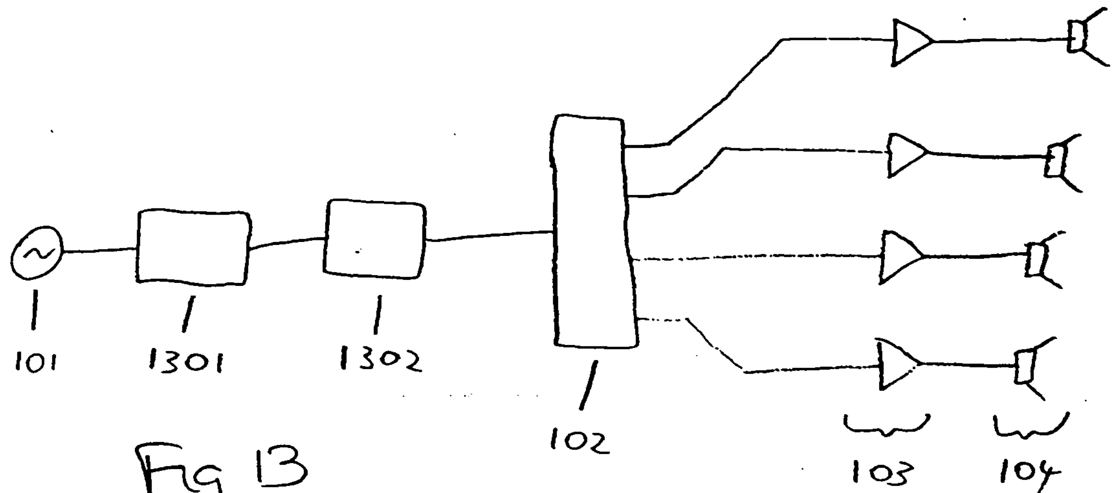

Fig 13

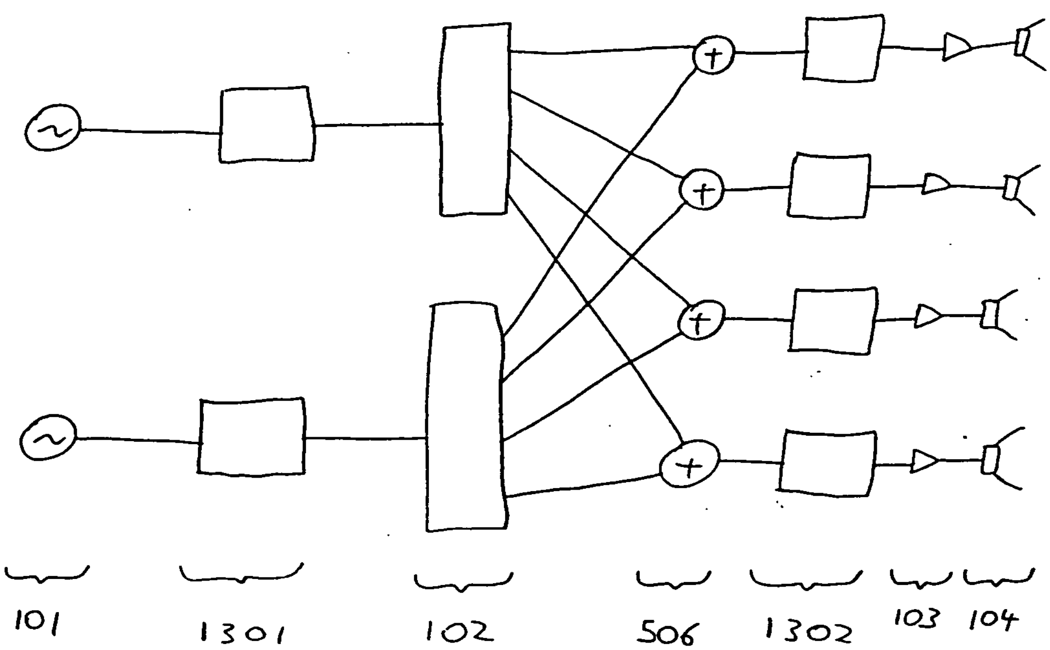

Fig 14

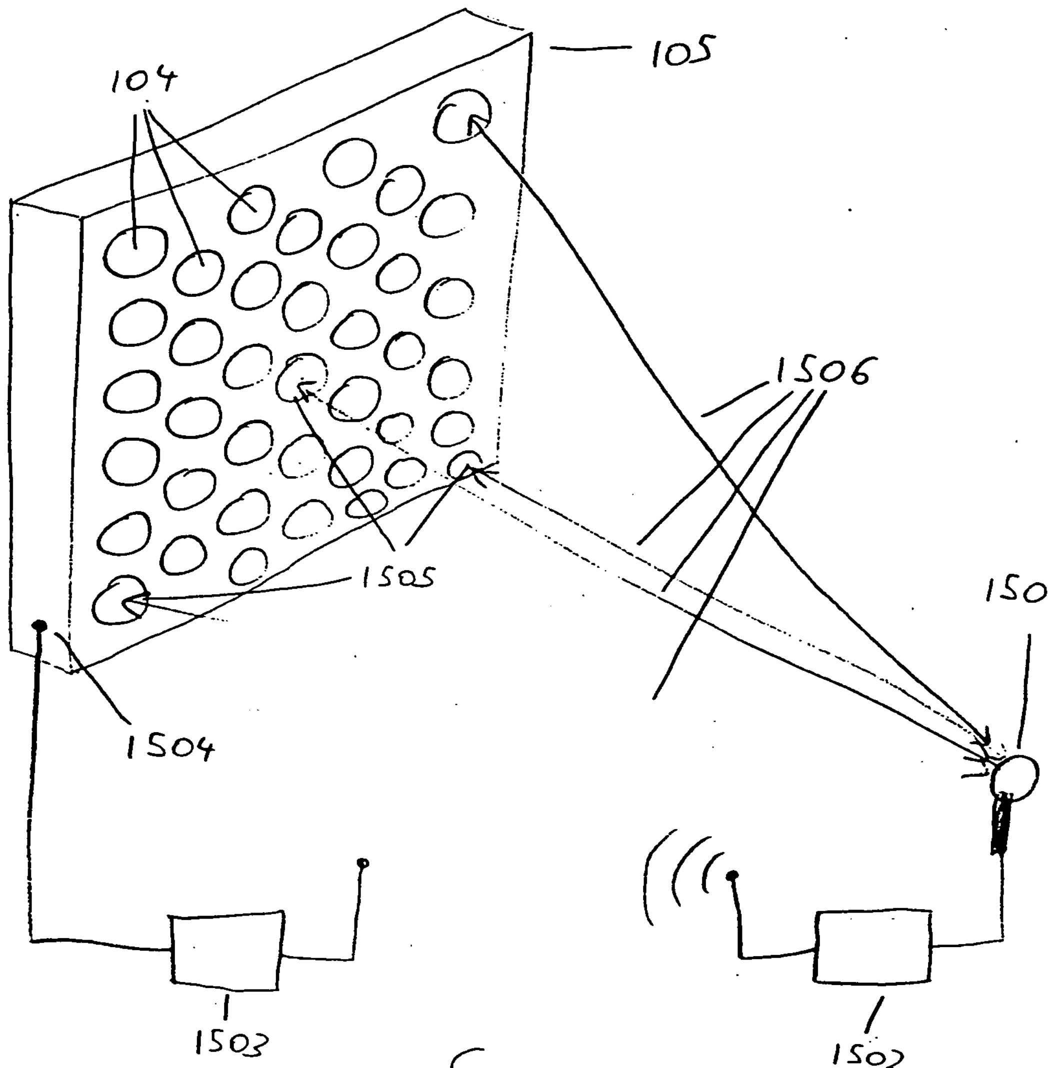

fig 15

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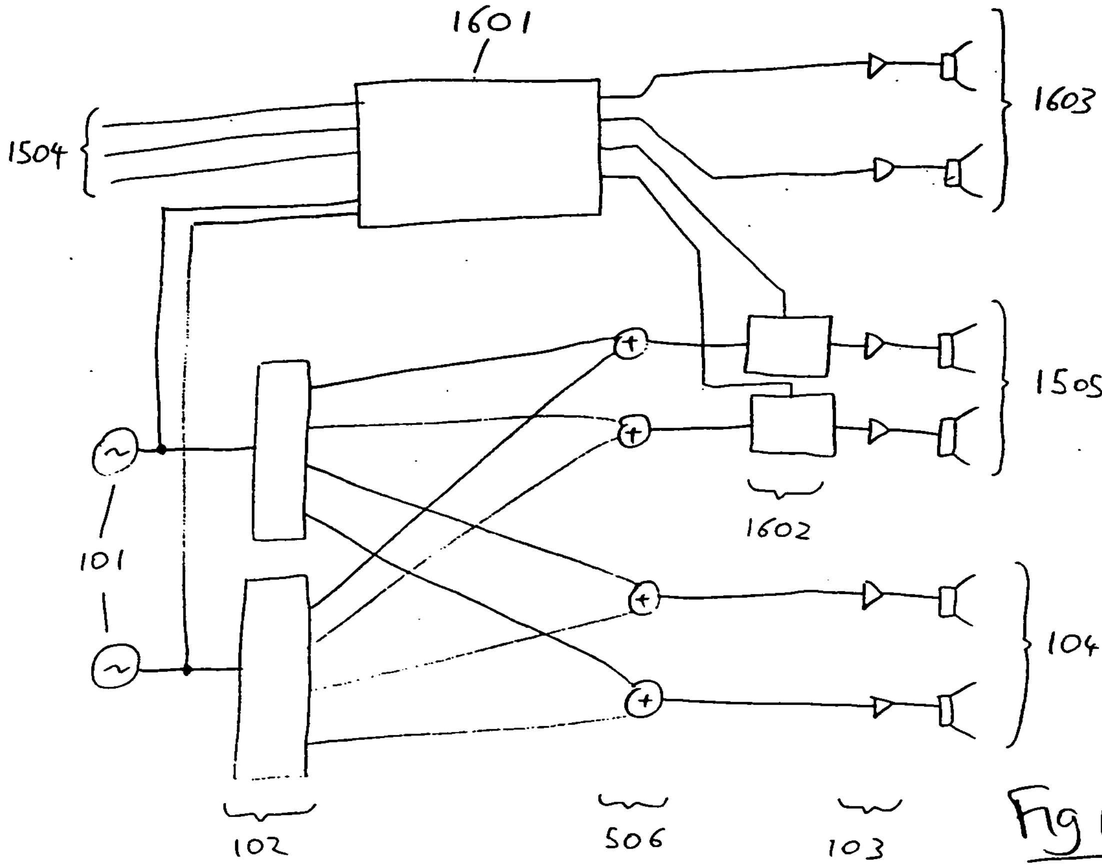
1601
1504
1603
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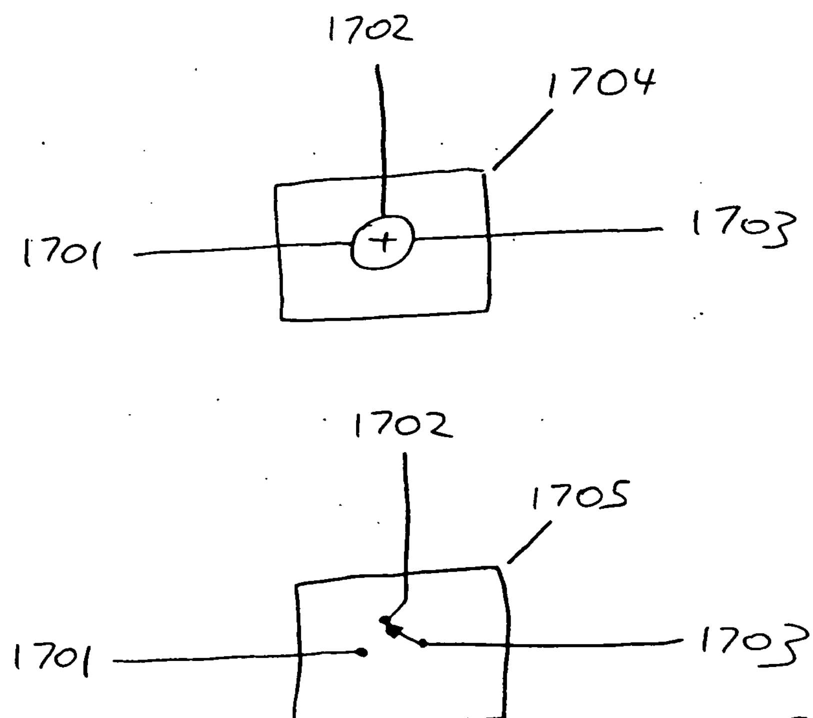
1602
101
104
102
506
103

Fig 16


1702
1704
1701
1703
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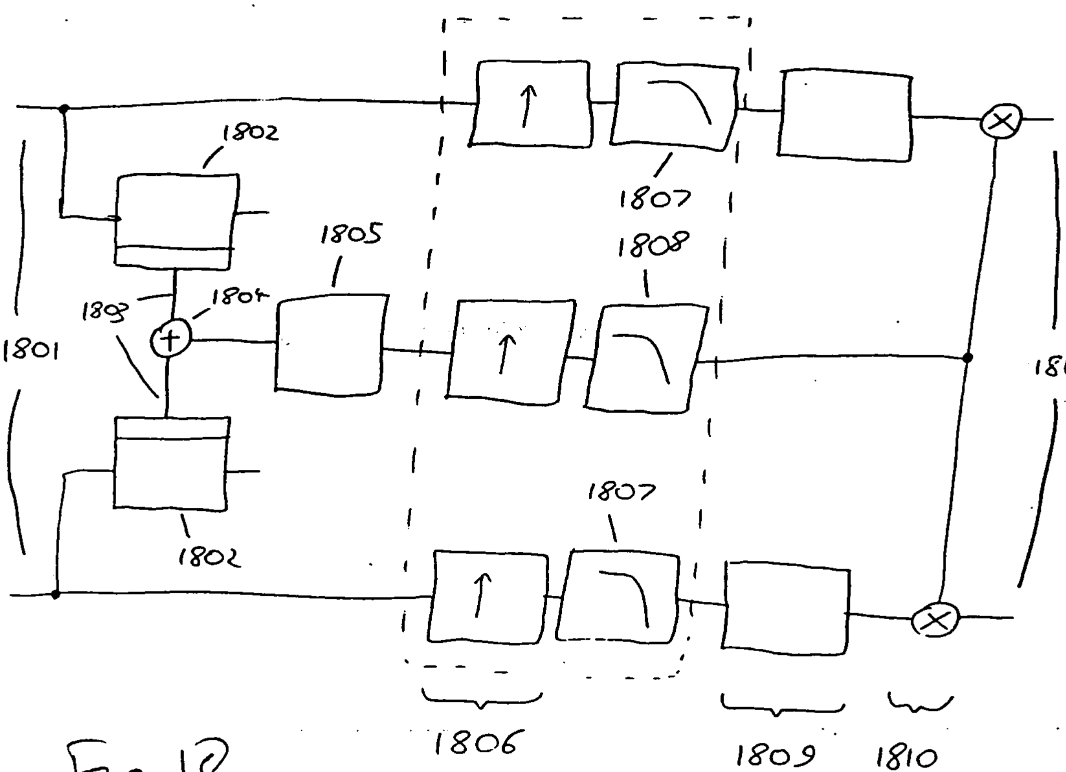
1705
1701
1703

Fig 17



1802
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1802
1806
1809
1810

Fig 18

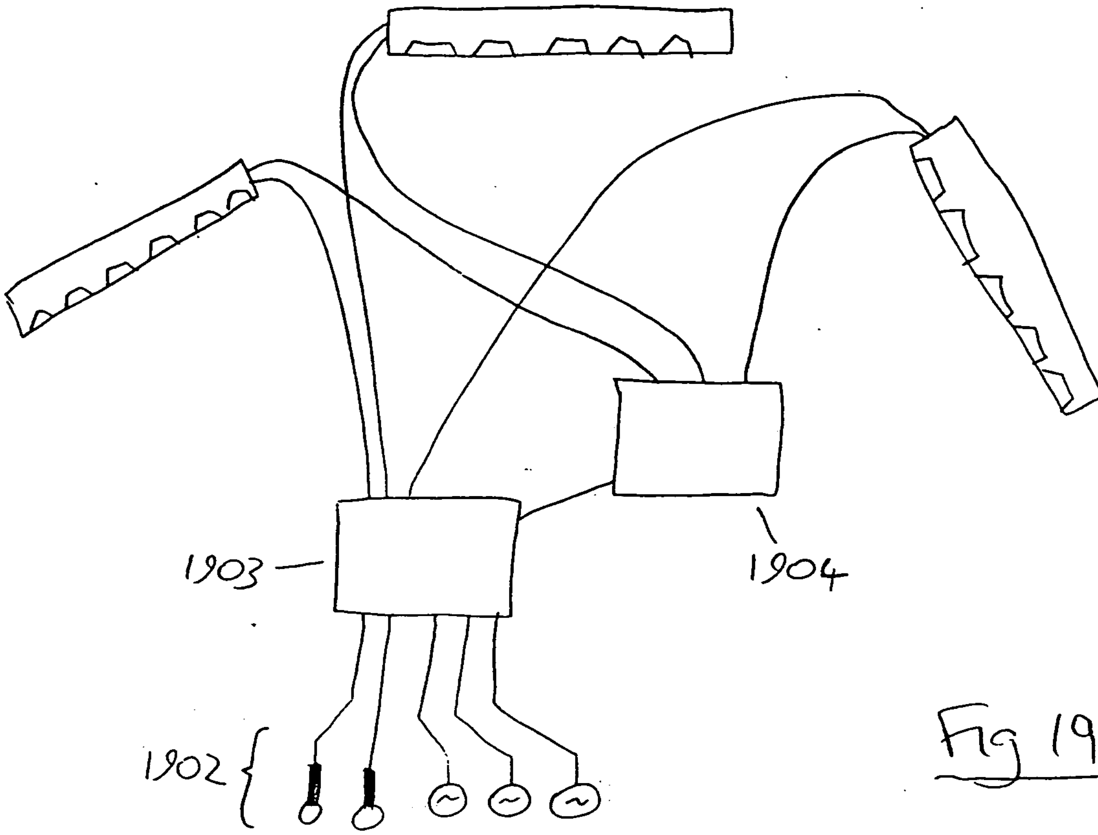

Fig 19

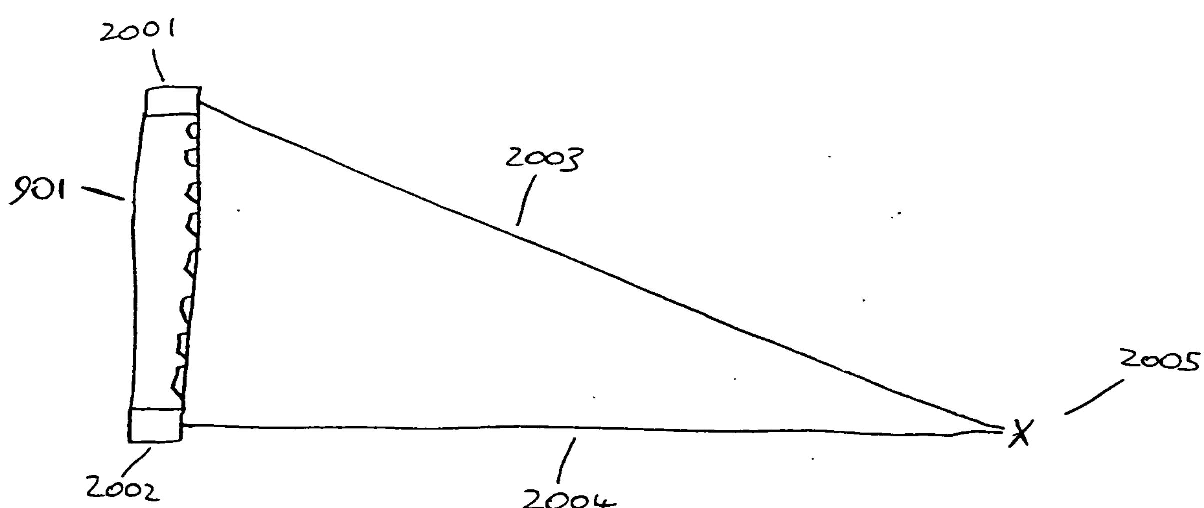

Fig 20

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